CHOICE OF SAMPLED WAVES FOR TIME-DIVISION MULTIPLEX TELEPHONE SYSTEMS

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ABSTRACT. We can have multiplex telephone systems by using sampled voice signals on time division basis. These sampled voice signals may be used to operate the telephone receivers directly. The paper deals with the choice of the optimum frequency and width at which the voice signals are to be sampled in order that the telephone receivers may give the maximum output without having any appreciable distortion.

INTRODUCTION

Recently time-division principles have been applied to multiplex radio relay systems (Grieg and Levine, 1946) and pulse code modulation methods have been used in many of them (Black, 1947 and Fieldman 1948) The use of sampled waves in time-division working has some advantages in telephone systems and although their use in telephone systems has not yet been established, work is being carried on in this direction (Chattermole, 1958 and Price, 1958). Sampling frequencies of different values and of different width have been tried but no investigation appears to have been made with regard to the determination of the optimum values of the sampling frequency and its width. Sampled voice signals are just like pulse amplitude modulated signals but in them as the modulations are unidirectional, no demodulation is required and they can be used to operate the telephone receivers directly. An attempt has been made in this paper to determine the optimum values of the sampling frequency and its width when sampled voice signals are used to operate the telephone receivers directly.

TELEPHONE RECEIVER AND SAMPLED VOICE SIGNALS

In the case of sampled voice signals with square-topped pulses separated by spaces of no pulses, not only will the original voice signal be present but there will also be a very large number of beat frequencies formed by the frequencies of the original wave with the fundamental and odd harmonics of the sampling frequency When the sampling frequency is greater than twice the highest frequency component present in the voice signal, all the frequency components of the voice signal will be present in the sampled wave according to Shanon's sampling theorem, but when a telephone receiver is subjected to such sampled voice signals, it will be subjected not only to the original voice signal but also to a very large number of beat frequency components referred to above. It is, therefore, clear that if the telephone receiver is to give an exact reproduction of a voice signal from a sampled wave, the sampling frequency should not only be greater than double the highest frequency component present in the voice signal, but the sampling frequency should be such that none of the other wave packets associated with sampled wave may affect the telephone receiver.

When the number of channels to be worked on sampled voice signals on time-division basis is increased, the duration of the samples are to be necessarily decreased in proportion. Thus for a 100 channel system, the duration of the samples must at least be decreased to 1/100th part of the time between consecutive samples. When the duration of the samples is very small, however compared to the time between consecutive samples, the telephone receiver may be assumed to be subjected to a series of impulses at the sampling points and the diaphragm may be assumed to be displaced to distances proportional to the total sum of the amplitudes of the impulses. In this case also the frequencies in the signal will be reproduced exactly when the sampling frequency is an exact multiple of them. This can be shown very easily mathematically. When they are not exact multiples, the sum total of the amplitudes due to the same frequency component will be different in different half cycles and sub-harmonics of the frequencies will be introduced. These have been illustrated in Figs 1(a), 1(b), 1(c) and 1(d) for a few cases. As the voice signal consists of a band of frequencies, any one sampling frequency cannot be exact multiple of all the components. Therefore distortions are liable to be introduced. This distortion, however, decreases as the sampling frequency is increased and it is negligible when the sampling frequency is many times the highest frequency component present in the voice signal.

MATHEMATICAL TREATMENT

Suppose p is the angular frequency of the original wave and W is the angular frequency of the sampling rate.

Further, let $yW = xp$ where y and x are the two minimum possible integral numbers.

Then the angular distance between two consecutive samples will be given by

$$\theta = 2\pi \frac{y}{x}$$

If m samples correspond to n cycles of the wave

Then $\theta.m. = n.2\pi$ (m and n are integral numbers)

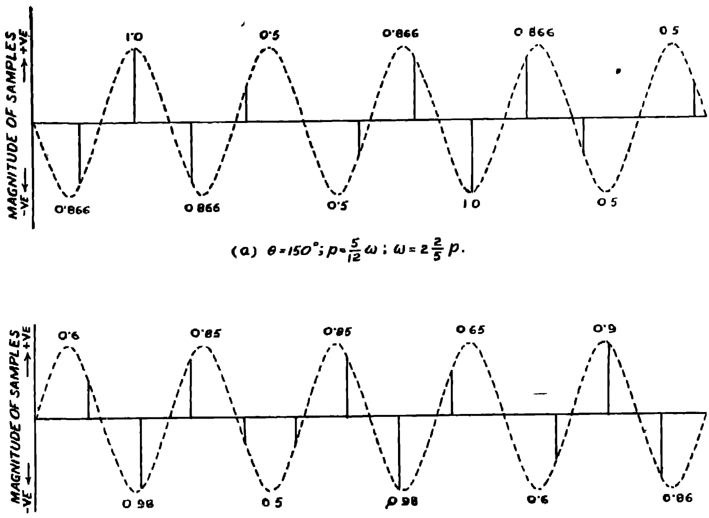
or
$$n = \frac{\theta.m.}{2\pi} = \frac{2\pi y}{x}\frac{m}{2\pi} = \frac{y}{x}\ .\ m$$

Thus for given values of y/x, m will be of such value that n becomes an integer.

When $W = 2p$, $3p$, $4p$ etc, n will become equal to 1. Therefore in each cycle there will be a definite number of samples and it can be shown that the total magnitude of the samples in each half portion is also equal. Therefore if a telephone receiver be subjected to such sampled waves instead of the continuous wave, it will give the same reproduction.

When, however, y/x has a value such that y has got a value other than 1, n will also have a value other than 1. In such a case a fixed number of samples cannot be contained in each cycle of the wave and so the samples in each cycle will be differently distributed but a definite number of samples will be repeated after every few and definite number of cycles of the wave, this number of cycles being given by the value of n. Further, the sum total magnitude of the samples in each half cycle will not be of the same value and a sub harmonic of the order of the value n will be introduced.

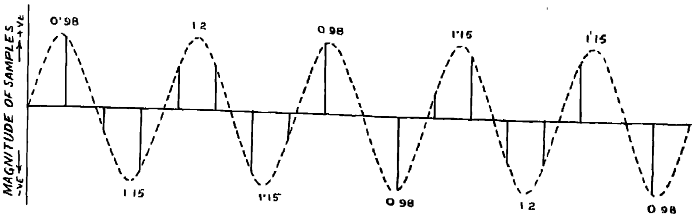
If a band of frequencies, say, voice frequencies are, therefore, sampled and a receiver is subjected to such a sampled wave, a few frequency components whose direct multiples will be the sampling frequency, will be correctly reproduced and others will be distorted. It is to be noted, however, that higher the values of sampling rates, greater will be the number of samples present in each half cycle of each component and less will be the differences in total magnitude of the samples. These will be evident from Figs. 1(a), 1(b) 1(c) and 1(d). Therefore, for good



(a) $\theta = 150°$; $p = \frac{5}{12}\omega$; $\omega = 2\frac{2}{5}p$.

(b) $\theta = 140°$; $p = \frac{7}{8}\omega$; $\omega = 2\frac{4}{7}p$.

Fig. 1(a) & 1(b). Total magnitude of pulses in different half cycles when sampling done at a frequency which is not an exact multiple of the frequency of the wave.



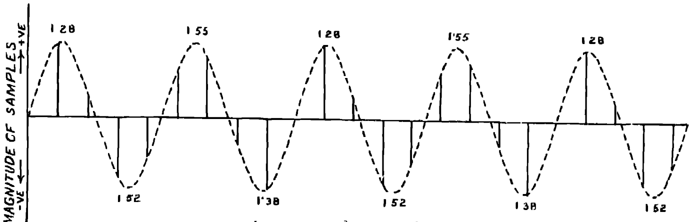


Fig. 2(c) & (d). Total magnitude of pulses in different half cycles when sampling is done at a frequency which is not an exact multiple of the frequency of the wave.

reproduction purposes, the sampling frequency should be quite a number of times the highest frequency component present in the voice signal. As discussed afterwards, it has been experimentally found that the sampling frequency should be at least 4 times the highest frequency component present in the voice signal, in order that the distortions in the output of the telephone receiver may be negligible.

METHODS OF PRODUCING SUITABLE SAMPLED WAVES

Germanium diodes in the form of a bridge have been used by the author elsewhere (Das, 1957) in producing sampled waves as shown in Fig. 2. The bridge shows a low resistance between the points A and B when C is at a higher potential than D and it shows a very high resistance when C is at a lower potential than D. If an alternating voltage referred to as the switching voltage be applied between C and D, then the bridge will be made conducting and non-conducting alternately between the points A and B and the bridge will behave like a switch between A and B. If pulses are used as switching voltage in series with a bias voltage as shown, then also the bridge will be conducting during pulse periods only if the pulse voltage is greater than the bias voltage. Pulses of varying frequency and varying width are obtained by triggering a monostable multivibrator

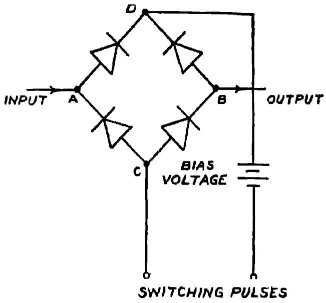


Fig. 2. Bride switching circuit with Bias voltage.

with a blocking oscillator. The frequency of pulses is changed by changing the frequency of triggering pulses with the help of the blocking oscillator and the duration of the pulses is changed with the help of the monostable multivibrator used. Thus using such pulses as switching voltages, we can get the sampled waves of different sampling frequency and of different width.

EXPERIMENTAL

(*i*) *Optimum sampling frequency*

The experimental arrangement is shown in Fig. 3. The switching voltage obtained in the way described above is applied to the bridge circuit consisting of four germinium diodes in series with a bias battery voltage of 5 volts. The audio signal was obtained from an audio oscillator and the sampled

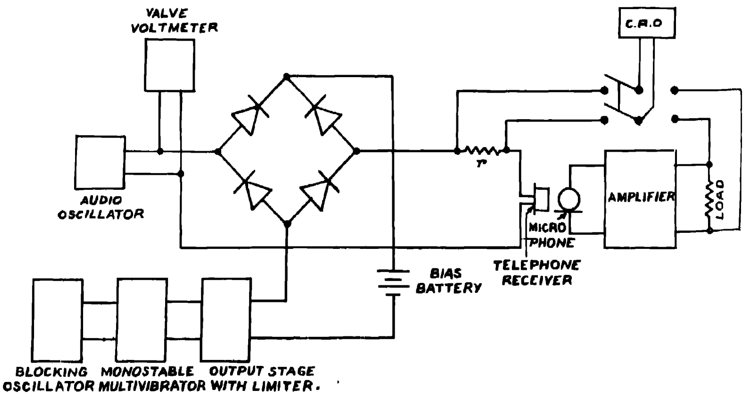
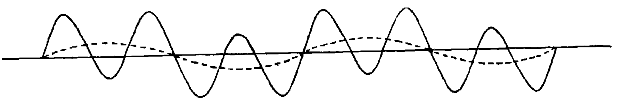


Fig. 3. Experimental arrangement.

wave obtained was connected directly to a telephone receiver The output of the telephone receiver is fed to a microphone and the output signal from the microphone is amplified by means of an amplifier whose output was studied with a C.R O. When the sampling frequency was above 10 Kc/s, the output of the telephone receiver was found to be an exact reproduction of the audio signal over the entire voice signal range up to 3 Kc/s irrespective of the width of the samples When the sampling frequency was lower than 10Kc/s, but an exact multiple of the audio signal, then also the output was found to be an exact reproduction of the signal which was sampled. When, however, the sampling frequency was less than 10Kc/s and not an exact multiple of the audio signal frequency, distortions were found to be present in the output and these distortions varied as the sampling frequency was changed evidently due to the different sub-harmonics that were present in the different cases. One such distorted signal as seen in C R.O. for a sinusoidal wave is shown in Fig. 4(a). When, however, the sampling frequency was 10 Kc/s and above, the output of the same audio wave as seen in C.R.O is as shown in Fig. 4(b)

(a)



(b)

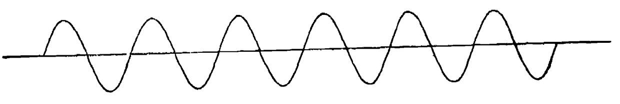


Fig. 4. (a) Output of a sinusoidal signal when sampling frequency is not an exact multiple of the signal and much below 10 Kc/s.

(b). Output of the same sinusoidal signal when the sampling frequency is 10 Kc/s.

(ii) Optimum width of samples

The power of the original wave is theoretically proportional to t^2/T^2 in the sampled wave where t is the width of the sample and T is the time interval between consecutive samples. Therefore, if keeping the sampling frequency constant, its width is decreased, the power of the original wave in the sampled wave will also be decreased and in order to get the same power output the power of the original wave has to be increased before it is sampled. This has been experimentally determined with the same experimental arrangement, with the slight modification by which the output can be measured. The results obtained are shown in Fig. 5. Excess power in *db* required in audio signals of 1000 c.p.s. before it is sampled in order that the sampled waves may give the same audio signal output in a telephone receiver, has been plotted against width of samples of different sampling frequency. It is seen that smaller the width of the samples, larger the power required in the audio signals for giving the same output from

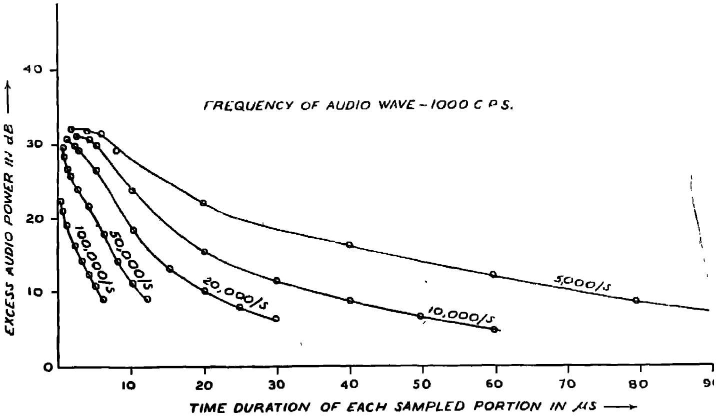


Fig. 5.

the sampled waves. The maximum width of samples is, however, determined by the sampling frequency and the number of channels in the system. With 10 Kc/s as the sampling frequency, the value of T is fixed i e. 100 μ-secs. Thus for a 10-channel system, the maximum theoretical value of the width of the sample can be 10 μ-secs. If we are to give a margin of 100 p c. distortion, the width of the samples will be limited to 5 μ-secs only. With such samples the audio signal has to be increased in power by about 30 db before sampling if the telephone receiver is to give the same output as the original audio signal.

DISCUSSION

When a telephone receiver is directly operated by sampled voice signals, the sampling frequency having a value equal to double the highest frequency component present in voice signals, the reproduction is not satisfactory because of the beat frequency components formed by the fundamental and odd harmonics of sampling frequency. For instance, if the sampling frequency is three times a particular frequency component in the voice signal, then the second harmonic will be introduced by the lower beat. For this reason although 3 Kc/s may be taken as the highest frequency component of the voice signals, a sampling frequency of 6 Kc/s does not reproduce the original signal faithfully. The beat frequencies formed by higher harmonics present in the square-topped signals may be negligible but the beat frequency formed by the fundamental of the sampling frequency cannot be neglected. An ordinary telephone receiver is not very sensitivea bove 7 Kc/s. Hence if the beat frequency formed by any component of the voice signal with at least the fundamental of the sampling frequency does not come within 7 Kc/s, the reproduction is not distorted. As the highest frequency component present in voice signals may be taken as 3 Kc/s, the lowest sampling frequency necessary is 10 Kc/s as experimentally determined

When the samples are of very short durations, distortion due to the sampling frequency being not an exact multiple of any voice signal frequency, is also negligible when the samling frequency is at least 10 Kc/s. The maximum width of the samples that can be used is limited by the number of channels for a particular sampling frequency. It is to be noted, however, that when the number of channels is increased and the width is necessarily decreased it is not necessary to increase the power of the audio signal proportionately as shown by the flat nature of curves of Fig 6 towards the smaller width regions. As it is desirable to use the lowest value of sampling frequency, 10 Kc/s is the optimum value for sampling voice signals to be used for working telephone receivers and the optimum width of the samples is determined by the number of channels in the system

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